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#7 / Appeal Brief
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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE BOARD OF APPEALS AND INTERFERENCES

IN THE APPLICATION OF	:	Keane, Michael
SERIAL NO	:	09/881,441
FILED	:	June 14, 2001
FOR	:	Measuring Speech Quality
EXAMINER	:	Harper, Vincent P
GROUP ART UNIT	:	2654
DOCKET NO	:	476-2037

I hereby certify that this correspondence is being deposited with the United States Postal Service as first class mail in an envelope addressed to "Director of Patents and Trademarks, Washington, D.C .20231," on October 25, 2002.

Name of person signing Jennifer J. ~~Ramirez~~

Signature

BRIEF ON APPEAL

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Honorable Director of Patents
and Trademarks
Washington, DC 20231

Dear Sir,

This appeal is from the Examiner's final rejection of September 6, 2002 in which claims 1 to 23 were rejected. A timely Notice of Appeal is filed herewith with the appropriate fee of $320.

This brief is being filed in triplicate, along with the required $320 fee pursuant to 37 C.F.R. § 1.17(c).

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(1) Real Party in Interest

This application is assigned to Nortel Networks Limited. The assignment is recorded at Reel 011905, Frame 0737.

(2) Related Appeals and Interferences

There are no related appeals or interferences.

(3) Status of Claims

This application was filed with claims 1 through 23, which have been finally rejected by the Examiner. No claims have been cancelled, and the rejection of claims 1 through 23 is appealed. Claims 1 through 23, as amended during the prosecution of the application, are set forth in the appendix.

(4) Status of Amendments

An amendment filed on July 10, 2002 was not deemed persuasive by the Examiner. The Examiner issued a Final Office Action on September 6, 2002 stating the identical reasons for rejecting claims 1 through 23 as in the previous Office Action of April 10, 2002. No amendments have been made following the Final Office Action.

(5) Summary of Invention

The present invention relates to a method and apparatus for measuring speech quality of a voice call made over a packet-based communications network. For example, the voice call may be a voice over Internet protocol (VoIP) call or any other suitable type of voice call made over a packet-based network.

During a VoIP telephone call, the voice signal from a user is processed by a digital signal processor and then compressed before being stored in packets that are suitable for being transported using internet protocol in compliance with one of the specifications for transmitting multimedia (voice, video, fax and data) across a communications network. The packets are transmitted across a communications network to a called party for example, using real time transport protocol (RTP).

When the packets are received at their destination, the voice signal is decompressed before being played to a called party. The specific path that the packets take over the communications network is not specified and can be any suitable path that is available. Thus, several different VoIP calls between the same destinations may take different actual paths over the communications network.

Any suitable compression/decompression scheme is used, and these are referred to as coder-decoder compression schemes (CODECs).

Previous methods of measuring speech quality have involved sending test speech signals in a dedicated voice call. For example, when the PESQ or other similar known algorithms are used to measure speech quality, a dedicated voice call is set up to transmit only test speech signals over a communications network. This enables the test voice signals to be easily identified and provides a means of determining the amount of degradation that occurs as a result of transmission over the network. However, it is known that network parameters such as packet loss and packet delay are significantly variable for many packet switched networks. Therefore, the results from a single test call over a packet switched network cannot be assumed to reflect the speech quality between the end-points on another occasion. This is especially because different voice calls between the same destinations on a packet-based network may take different actual paths over the communications network. This means that the test packets may be degraded as a result of transmission through the network in a different manner than the packets of the ongoing voice call.

The present invention recognizes these problems and solves them by adding test voice information to voice call packets of an ongoing voice call. Then when the packets are received at a node in the network the transmitted test voice information is extracted and compared with a pre-stored copy of the test voice information. Because the test voice information is added to voice call packets of an ongoing voice call, it experiences the same degradation as the voice information itself. In this way, a more accurate measure of speech quality for the voice call is obtained than

has been possible in the previous methods that used dedicated voice calls to transmit only test speech signals.

As explained in the response of July 10, 2002, the principal difference between the present invention and the prior art is that the same stored test voice information is stored at every node or terminal where a speech quality measure might be required and that voice and test information are carried on the same call. The reason that this is so important is as follows:

by having stored data test voice information at each node terminal all users have an exact copy of something with which to make a comparison;

introduction of the test voice information with a real voice payload ensures that on transmission to another node terminal the test voice information and the voice suffer the same effects which means that the speech quality measure is useful;

on reception at the other node or terminal a comparison between the stored test voice information at that node or terminal and the test voice information which has been transmitted gives an exact measure of the impact of the network on speech quality.

(6) Issues

Seven issues are presented:

Claim 11 is rejected under U.S.C. § 102 as allegedly being anticipated by Lewis et al. 6,330,428.

Claims 1, 3, 7, 8, 14, 16, 17, 18, 19, 20 and 21 are rejected under 35 U.S.C. 103(a) as allegedly being unpatentable over Lewis in view of Tschudin ("Header hopping and packet mixers," Ninth Conference on Computer Communications and Networks, 2000. Proceedings, Oct 2000).

Claims 4, 9, 10, 22 and 23 are rejected under 35 U.S.C. 103(a) as being allegedly unpatentable over Lewis in view of Tschudin, and further in view of well-known prior art.

Claims 2, 5 and 15 are rejected under 35 U.S.C. 103(a) as allegedly being unpatentable over Lewis in view of Tschudin and further in view of Petitcolas et al., ("Information Hiding – A Survey" Proceedings of the IEEE, Vol. 87, No. 7, July 1999).

Claim 6 is rejected under 35 U.S.C. 103(a) as allegedly being unpatentable over Lewis in view of Tschudin and Petitcolas, and further in view of well-known prior art.

Claim 12 is rejected under 35 U.S.C. 103(a) as being allegedly unpatentable over Lewis in view of Tschudin and Petitcolas.

Claim 13 is rejected under 35 U.S.C.103(a) as being allegedly unpatentable over Lewis in view of well-known prior art.

(7) Grouping of Claims

The claims do not stand or fall together.

(8) Argument

Claim 11 under U.S.C. §102 as allegedly being anticipated by Lewis et al. 6,330,428.

As explained in the response of July 10, 2002, Lewis does not teach providing each node or terminal with the same stored test voice information and sending test data and voice data embedded together on the same call.

Lewis teaches a system which sends voice and test data on different calls. This means it is likely that the voice and test data do not travel the same way and as such do not suffer the same impacts from the network. There is no stored test voice

information at each node/terminal in the network. As such Lewis cannot give an accurate measure of the speech quality of a specific call in the network and suffers from many of the drawbacks identified in the introduction of the present application.

The Examiner argues that the test voice signal of Lewis (column 3, lines 2-3; column 3, lines 18-21) is a signal for a voice call as claimed in claim 11. This is not the case. The test voice signal of Lewis is independent of and separate from an actual voice call. For example, Lewis explains that a voice sample from memory is used (column 3, line 1) which would not be the case for an actual live voice call. The test voice signal of Lewis does not comprise any actual speech signals for a communication session between a calling and a called party as is the case is a voice call. For example, column 6 lines 8 to 18 of Lewis explain how the voice samples are taken from 20 speakers and stored.

<u>Claims 1, 3, 7, 8, 14, 16, 17, 18, 19, 20 and 21 as allegedly being unpatentable over Lewis in view of Tschudin.</u>

Regarding independent claim 1, the Examiner argues that Lewis discloses all the features of this claim except for "adding at least part of the stored test voice information to at least some of the packets". The Examiner then goes on to argue that this feature is taught in Tschudin. Both these arguments are respectfully traversed.

As explained above, Lewis does not teach providing each node or terminal with the same stored test voice information and sending test data and voice data embedded together on the same line voice call. In contrast, Lewis uses a dedicated test call.

The Examiner states that the first and second nodes of claim 1 read onto the voice terminals 154, 158 of Lewis' Figure 1B. However these nodes do not each comprise the same stored test voice information as specified in claim 1. As described in Lewis column 5 lines 6 to 22 and column 5 lines 50 to 62, it is the VQPE which stores the test voice samples and which carries out the comparison. The mobile unit 104 codes the test voice samples, transmits them to the wireless network 110, receives

them back from the wireless network 110, decodes them, and sends them back to the VQPE. Only the VQPE stores the test voice samples.

The Examiner goes on to argue that one of the voice terminals 154, 158 receives an original test voice sample and that this reads onto Applicant's claim 1 step (i) of, at the first node, receiving packets for the voice call. This is not the case because the voice terminal in Lewis is receiving test voice samples, not packets for an actual voice call.

The Examiner also states that Lewis discloses step (iii) of Applicant's claim 1 which specifies that at the second node a comparison is made using a speech quality assessment algorithm. This is not the case because in Lewis the VQPE carries out a comparison. The VQPE is not the second node because the Examiner argues that is the second voice terminal.

Tschudin describes a method of making communications confidential by hiding or embedding those communications in other "dummy" data. That is Tschudin describes a method to provide confidentially without encryption, by introducing "chaff" (fake packets) into the true packet sequence (the "wheat"). These packets are introduced based on a pseudo random sequence, with this sequence known only to the sender and the receiver. These packets are not tagged in a real sense, but identifiable via the pseudo random sequence (this could only be considered virtual tagging).

The Examiner states that it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify Lewis to allow testing and normal voice communication to proceed simultaneously by the use of steganographic techniques, for the purpose of having an ongoing evaluation of speech quality. This is respectfully traversed.

Lewis does not teach providing each node or terminal with the same stored test voice information and sending test data and voice data embedded together on the

same call. There is nothing in Lewis to direct the skilled person to modify Lewis to provide those features. That is, starting from Lewis, the skilled person has a voice quality performance evaluator for assessing the performance of a mobile unit or a communications network. There is nothing in Lewis to have made the skilled person want to do anything different from Lewis. Lewis does not recognize the problems that dedicated test calls give rise to in contrast to the present application, which recognizes and is specifically concerned with these problems.

Even if the skilled person had combined Lewis and Tschudin he would not have reached the invention as claimed in claim 1. Neither Lewis nor Tschudin describe providing each node or terminal with the same stored test voice information as specified in claim 1. Also, Tschudin describes introducing "chaff" (Fake products) into the "wheat" (true packet sequence). This is not the same as embedding test data and voice data together on the same call, neither of which are "chaff" or fake packets. Tschudin's aim or purpose is to provide confidentiality and the skilled person would not have considered Tschudin because confidentiality is unrelated to the problems of the present invention.

Claims 3, 7 and 8 are dependent upon claim 1 and are therefore patentable over Lewis and Tschudin for the same reasons as given for claim 1 above.

Regarding independent claims 14, 16, 17, 18, 19, 20 and 21 these all include at least some features corresponding to those of claim 1 and are therefore patentable over Lewis in view of Tschudin for the same reasons as given above for claim 1.

<u>Claims 4, 9, 10, 22 and 23 as allegedly being unpatentable over Lewis in view of Tschudin and well known prior art.</u>

The Examiner's arguments here rely on the assumption that the independent claims from which these claims depend are not patentable over Lewis in view of Tschudin. However, this is not the case for the reasons given above. There is no motivation for the skilled person to modify Lewis, let alone to modify Lewis by adding features from Tschudin, providing each node with stored test voice information and also,

making additional modifications that the Examiner alleges are well known in the art. No evidence is provided by the Examiner that such features were in fact well known in the art at the time of the invention.

Claims 2, 5 and 15 as allegedly being unpatentable over Lewis in view of Tschudin and further in view of Petitcolas.

The Examiner's arguments here rely on the assumption that the independent claims from which these claims depend are not patentable over Lewis in view of Tschudin. However, this is not the case for the reasons given above.

Claims 2 and 15 are directed to an embodiment of the invention in which silent periods during a voice call are identified and test voice information is added to the voice call packets during these silent periods. This is advantageous because a live voice call is not disrupted or otherwise adversely affected by addition of the test voice information. (See page 6 paragraph 4 of the specification.) The Examiner acknowledges that neither Lewis nor Tschudin describe this feature. The Examiner contends that the concept of identifying packets where speech is absent is consistent with the technique of identifying residual bandwidth for the purpose of storing hidden information there, as taught by Petitcolas. However, this is not the case. The aim in the present invention is to send test voice information as part of an ongoing voice call in order that both types of information experience the same effects. Petitcolas is concerned with information hiding for confidentiality and security. The skilled artisan would not have been motivated to consider Petitcolas because of this. Also, page 1067 section D of Petitcolas that the Examiner directs one to describes adding information to be hidden to a cover object or image. The added information must necessarily disrupt the cover object or image to some extent. This is not the case in the present invention where test information is only added during "silent periods" specifically with the aim of not disrupting the voice call. For these reasons, even if the skilled person had combined Lewis, Tschudin and Petitcolas he would not have reached the present invention as specified in claims 2 and 15.

Claim 6 as allegedly being unpatentable over Lewis in view of Tschudin and Petitcolas and further in view of well known prior art.

The Examiner's arguments here rely on the assumption that the independent claim from which claim 6 depends is not patentable over Lewis in view of Tschudin. However, this is not the case for the reasons given above.

In addition, it is submitted that the skilled person would not have been able to combine as many as four independent pieces of information as proposed by the Examiner without some degree of ingenuity and inventiveness, even if he had been motivated to do so, which is not the case and not explained by the Examiner. The Examiner has used perfect hindsight, employing the present invention as the means of mosaicing the prior art. That, of course, is improper.

Claim 12 as allegedly being unpatentable over Lewis in view of Tschudin and Petitcolas.

Claim 12 is dependent on claim 11 and claim 11 is patentable over Lewis for the reasons given above. The feature of claim 12 is similar to that of claims 2 and 15 and therefore the arguments made above in connection with claims 2 and 15 also apply here. For these reasons claim 12 is patentable.

Claim 13 as allegedly being unpatentable over Lewis in view of well known prior art.

Claim 13 is dependent on claim 11, which is patentable over Lewis for the reasons given above. Therefore claim 13 is patentable.

It is therefore submitted that the claims, as set forth in the Appendix, distinguish from, and are allowable over, the prior art. Reversal of the Examiner is in order and is therefore solicited.

October 25, 2002

Respectfully submitted,

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APPENDIX A

(Claims on Appeal)

1. A method of measuring the speech quality of a voice call between a first node and a second node in a packet-based communications network, each of the first and second nodes comprising the same stored test voice information, the method comprising the steps of, at the first node:

(i) receiving packets for the voice call and adding at least part of the stored test voice information to at least some of the packets;

(ii) forwarding the packets to the second node;

(iii) at the second node, accessing the stored test voice information at the second node and comparing it with the test voice information received in the packets using a speech quality assessment algorithm in order to obtain a measure of speech quality for the voice call.

2. A method as claimed in claim 1 wherein some of the packets received at the first node comprise voice information associated with the voice call and others of those packets are associated with periods when speech is absent from the voice call and wherein said step (i) further comprises identifying those packets which are associated with periods when speech is absent from the voice call and adding test voice information to one or more of those packets.

3. A method as claimed in claim 1 wherein said packet-based communications network is an Internet protocol communications network.

4. A method as claimed in claim 1 wherein said voice call comprises a real-time transport protocol session between the first and second nodes.

5. A method as claimed in claim 2 which further comprises making an indication in a header of each of those packets to which test voice information is added.

6. A method as claimed in claim 5 wherein said indication is a payload value and said packets are real-time transport protocol packets.

7. A method as claimed in claim 1 which further comprises, at the second node, identifying which of the packets comprise test voice information by determining whether a pre-specified identifier is present in a header of each of the packets.

8. A method as claimed in claim 7 wherein the packets are forwarded from the first node to the second node via one or more other nodes which do not have access to information about the pre-specified identifier.

9. A method as claimed in claim 1 wherein said first and second nodes are located substantially at the edge of the communications network.

10. A method as claimed in claim 1 wherein said speech quality assessment algorithm is a PESQ algorithm.

11. A signal for a voice call provided over a packet-based communications network, said signal comprising a plurality of packets at least some of which comprise test voice information for comparison as a node with stored test voice information which is the same as the test voice information.

12. A signal as claimed in claim 11 wherein some of the packets are associated with periods when speech is absent from the voice call and comprise test voice information.

13. A signal as claimed in claim 11 wherein the packets are real-time transport protocol packets and some of the packets comprise a header with an indicator, indicating that those packets comprise test voice information.

14. A packet-based communications network node arranged to enable speech quality to be measured for a voice call which is ongoing between a caller and a called party wherein the caller and the called party each have stored test voice information said node comprising:

(i) an input arranged to receive packets for the voice call; and

(ii) a processor arranged to add test voice information to one or more of the packets;

(iii) an output arranged to forward the packets towards the called party for comparison of the test voice information with the stored test voice information of the called party to provide a measure of said speech quality.

15. A network node as claimed in claim 14 wherein some of the packets received at the input comprise voice information associated with the voice call and others of those packets are associated with periods when speech is absent from the voice call and wherein the processor is further arranged to identify those packets which are associated with periods when speech is absent from the voice call and add test voice information to one or more of those packets.

16. A packet-based communications network node arranged to measure speech quality for a call which is ongoing between a caller and a called party, said node comprising:

(i) an input arranged to receive packets as part of the voice call some of which comprise voice information associated with the voice call and some of which comprise received test voice information;

(ii) stored test voice information;

(iii) a processor arranged to compare the received test voice information and the stored test voice information using a speech quality assessment algorithm in order to obtain a measure of speech quality for the voice call

17. A communications network comprising a first packet-based communications network node arranged to enable speech quality to be measured for a voice call which is ongoing between a caller and a called party wherein the caller and the called party each have stored test voice information, said node comprising:

(i) a first input arranged to receive packets for the voice call; and

(ii) a first processor arranged to add test voice information to one or more of the packets;

(iii) a first output arranged to forward the packets towards the called party for comparison of the test voice information with the stored test voice information of the called party to provide a measure of said speech quality;

and a second packet-based communications network node arranged to measure speech quality for a call which is ongoing between a caller and a called party, said node comprising:

(i) a second input arranged to receive packets as part of the voice call some of which comprise voice information associated with the voice call and some of which comprise received test voice information;

(ii) stored test voice information;

(iii) a second processor arranged to compare the received test voice information using a speech quality assessment algorithm in order to obtain the measure of speech quality for the voice call.

18. A method of measuring speech quality for a call which is ongoing, said method comprising, at a node in a packet based communications network:

(i) receiving packets as part of the voice call some of which comprise voice information associated with the voice call and some of which comprise received test voice information;

(ii) accessing stored test voice information at the node;

(iii) comparing the received test voice information and the accessed stored test voice information using a speech quality assessment algorithm in order to obtain a measure of speech quality for the voice call.

19. A method of enabling speech quality to be measured for a voice call which is ongoing between a caller and a called party said method comprising, at a node in a packet based communications network:

(i) receiving packets for the voice call;

(ii) adding test voice information to one or more of the packets; and

(iii) forwarding the packets towards the called party;

(iv) at the called party node extracting the received test voice information and comparing it with stored test voice information at said called party node to provide a measure of said speech quality.

20. A computer program for controlling a packet-based communications network node in order to enable speech quality to be measured for a voice call which is ongoing between a caller and a called party said computer program being arranged to control the node such that:

(i) packets for the voice call are received;

(ii) test voice information is added to one or more of the packets; and

(iii) the packets are forwarded towards the called party.

(iv) at the called party node the received test voice information is compared with stored test voice information at said called party node to provide a measure of said speech quality.

21. A computer program arranged to control a packet-based communications network node in order to measure speech quality for a call which is ongoing between a caller and a called party, said computer program being arranged to control the node such that:

(i) packets are received as part of the voice call some of which comprise voice information associated with the voice call and some of which comprise received test voice information;

(ii) stored test voice information at the node is accessed; and

(iii) the received test voice information and the stored test voice information are compared using a speech quality assessment algorithm in order to obtain a measure of speech quality for the voice call.

22. A computer program as claimed in claim 21 which is stored on a computer readable medium.

A computer program as claimed in claim 20 which is stored on a computer readable medium.